BBC

Research Department Report

DIGITAL VIDEO:
Studio signal processing

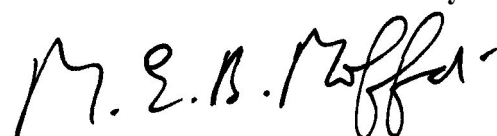**C.K.P. Clarke,** B.Sc.(Eng), A.C.G.I.

DIGITAL VIDEO : STUDIO SIGNAL PROCESSING

C.K.P. Clarke, B.Sc.(Eng), A.C.G.I.

Summary

Several features of the digital component standard for television studios are described with a view to explaining the principles on which they are based and the modes of operation which were intended. For example, the sampling and filtering techniques were chosen to support multiple signal conversions with minimal loss of quality and the multiplexing techniques used were selected to simplify interconnections. The implications of the coding parameters for signal processing and studio synchronisation are discussed and some simple test waveforms are suggested.

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1. INTRODUCTION

1.1 Signal coding for digital studios

It has been recognised for many years that there would be significant advantages in the use of digital techniques for television signal processing and distribution. Even so, it has only recently become possible to consider complete digitisation of the television system between the camera output and the broadcast transmitter. This has occurred partly through the general progress in digital integrated circuit technology and partly as a result of advances made in the recording of digital signals on magnetic tape.

For a considerable period, it was assumed that television signals would be digitised in composite PAL form[1] and digital systems based on composite PAL have been developed for signal distribution[2]. However, more recently it was recognised that the use of digital PAL signals throughout the network would place too many constraints on the operation of digital studios[3]. In particular, several important studio operations would require the signals to be decoded to component form in order to avoid the complication of processing modulated colour signals. Imperfections in the multiple PAL decode and recode operations involved would then have caused a rapid deterioration of picture quality.

The alternative approach was to work towards a system based on digital component signals, in which ultimately the signals would remain in component form throughout, only being encoded as PAL signals for broadcast transmission. With this approach, there would be a changeover period in which some PAL decoding and recoding would still be required[4,5], but the number of such processes would be relatively small. As the conversion to digital components operation progressed, the decoders and recoders would gradually be removed.

1.2 Standardisation of coding parameters

Interest in a European standard for digital component coding arose in response to the difficulties of exchanging programmes between broadcasters using 625-line, 50 field/sec (625/50) PAL and SECAM in the Eurovision network of the European Broadcasting Union (EBU). Component coding held the prospect that the signals would be identical whether sourced in a PAL or a SECAM country and could be encoded subsequently to the appropriate composite form for national broadcasting. When the considerable problems associated with processing digital composite PAL signals were appreciated, the discussions were broadened to include the general application of digital component signals to studio applications, in addition to international programme exchange.

Investigations associated with the EBU discussions occurred in two phases, each culminating in a series of demonstrations to the committees concerned. These sought to examine both the basic quality of potential standards and their suitability for signal processing, with experimental equipment being provided by several broadcasters. The development of BBC equipment in support of these demonstrations is described elsewhere[6].

From these investigations, and in consultation with the Society of Motion Picture and Television Engineers (SMPTE), the main parameters of the digital component signal coding standard for use in television production facilities[7] were chosen. Since then more detailed specifications, incorporating these basic parameters, have been agreed for bit-parallel and bit-serial interfaces[8] to provide for digital interconnections between items of studio equipment.

Although the digital coding standard is designed to suit a wide range of applications, its features reflect a number of assumptions made at the time about the way that digital equipment would be used, both initially and in the longer term. While the standard itself remains, some of the underlying assumptions may not be apparent. This Report therefore reviews some of the background, first by approaching the problem of introducing digital component signals into existing operational areas and examining the alternative techniques for obtaining digital component signals from analogue *RGB*. The remaining sections highlight features of the standard that allow for multiple conversions, for signal processing, for digital interconnections and for synchronisation. Also, the requirements for test signals are briefly considered. It is envisaged that the component coding approach will allow digital signals initially to co-exist with and eventually to supplant the present composite PAL signals, to provide greater operational flexibility and improved picture quality.

2. DIGITAL STUDIO DEVELOPMENTS

2.1 The changeover to digital studio processing

The complexity of studio installations dictates that an evolutionary approach to the introduction of digital component signals has to be followed. Because

of this, the new digital equipment will have to co-exist with the current analogue composite signals for a considerable period, initially forming 'islands' of digital components in an analogue composite 'sea'.

The earliest digital equipment was predominantly calculation intensive, for example standards converters, frame store effects systems and post-production processors. While digital component signals were used internally to avoid the complexity of processing colour subcarrier signals, each item included a composite decoder and recoder for operation with existing PAL signals, as shown in Fig. 1(a). As several such devices could be present in the signal chain, decoding and recoding might occur several times.

In view of the problems of decoding and recoding PAL, there has been a tendency to convert whole areas to digital operation[9] or to use digital equipment in the earliest part of the signal chain, before composite colour coding has taken place[10,11]. Thus, the analogue RGB signals from source devices are converted directly to digital component form, as shown in Fig. 1(b), or signals are made directly in component form by digital synthesis. Such installations have proved to be of great benefit for production flexibility and improved picture quality.

Analogue RGB signals have also been used as a convenient means of interconnecting digital component units in advance of the development of digital
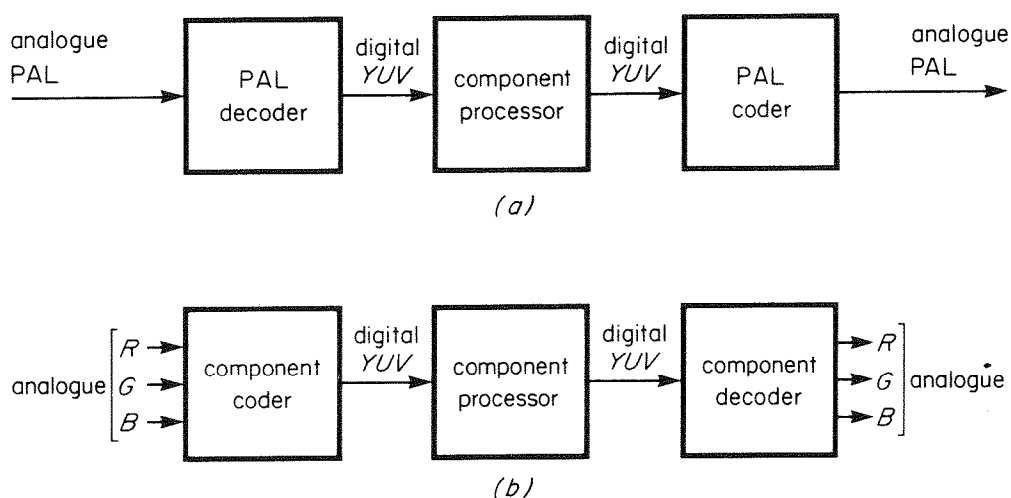

Fig. 1
Conversion interfaces between existing analogue signals and digital component signals: (a) with PAL and (b) with RGB.

Because chrominance and luminance share the same frequency band in the composite signal, it is generally impossible to separate the signals perfectly. When decoded, the viewed picture is subject to cross-effects in luminance and chrominance. In most circumstances, although this degrades the picture quality by a small amount, it is accepted as a minor penalty for obtaining compatible colour transmissions in a bandwidth originally set for monochrome signals. In a studio context, however, because the signals are recoded, the cross-effects from the first encoding process mix with those from the second and can lead to more serious impairments when the signal is finally decoded at the viewer's receiver. The impairments are such that decoding and recoding even once or twice can noticeably reduce the quality of the final picture. Furthermore, high quality decoders using comb filters for better separation of luminance and chrominance add significantly to the cost of the processing equipment. For these reasons, therefore, it is necessary to minimise the number of occasions on which a signal is converted between the composite and component forms. Techniques for PAL decoding and encoding in a digital component environment have been described in detail in an earlier Report[5].

interconnection and switching systems. This results in the possibility of converting between analogue RGB and digital components, as shown in Fig. 1(b), several times over. Unlike the conversions for PAL, these conversions can be accomplished with minimal signal impairment.

Perhaps the greatest challenge of the change to digital component working is to provide digital switching matrices and digital interconnections. This is necessary if significant advances are to be made beyond the current stage in the installation of digital component systems and to take full advantage of the capabilities of digital component video tape recorders. It may then be necessary to duplicate the signal routeing and switching functions in both analogue and digital forms, as shown in Fig. 2, for an extended period. This approach maintains operational flexibility and reduces the number of times that composite to component conversions would be needed by concentrating the equipment at a single point between the two matrices. As the source areas change to component signals, the digital matrix would be used increasingly, perhaps eventually replacing the composite matrix altogether.

PAL
sources

digital
sources

PAL
decoders

Fig. 2
Signal switching arrangements
for analogue PAL and digital
component signals during the
changeover period.

analogue
composite
matrix

digital
component
matrix

PAL
coders

PAL
outputs
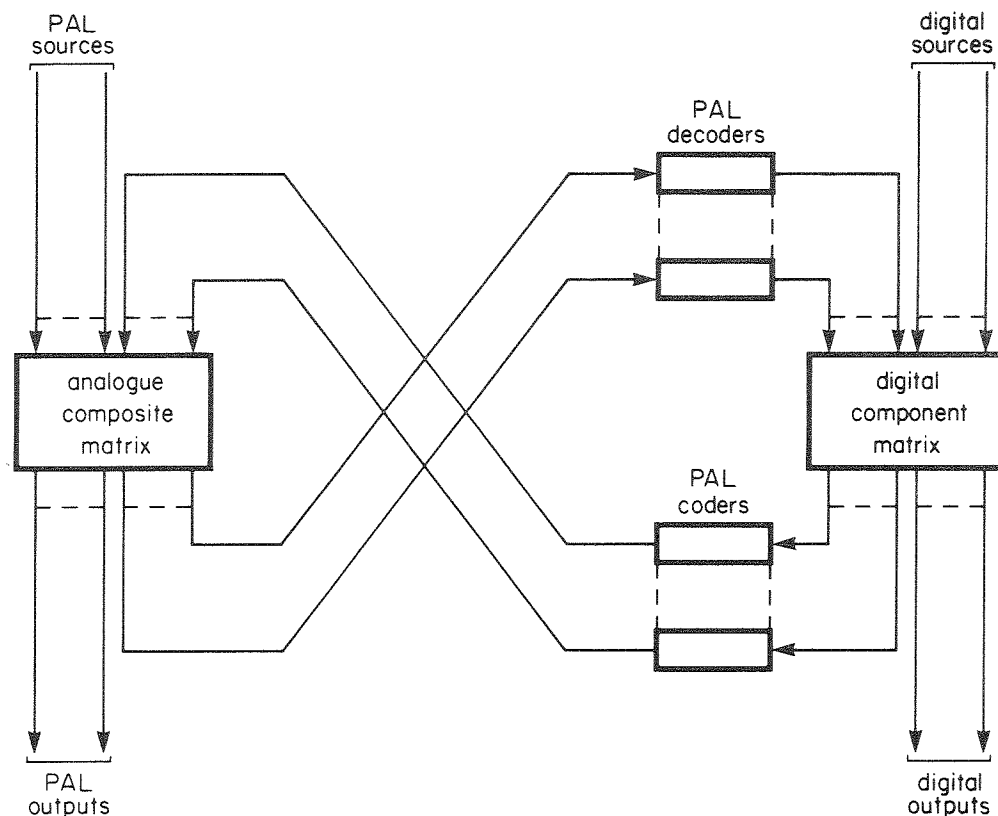
digital
outputs

2.2 Standards hierarchy

A particular feature of the CCIR Recommendation for digital component signal coding[7] is that it envisages the co-existence of a hierarchy of compatible coding standards. With the accepted notation, a digital component system with sampling frequencies of 13.5, 6.75 and 6.75 MHz for the luminance and two colour-difference signals, respectively, is denoted 4:2:2. As the digital interface standard[8], based on 4:2:2 sampling, is to be used to connect together major items of digital studio equipment, the 4:2:2 sampling parameters will be fundamental to any digital studio system.

Even so, the Recommendation allows for other coding standards, provided that conversion to 4:2:2 sampling is straightforward. In practice, this restricts the choice to sampling frequencies with simple relationships to 4:2:2 frequencies, such as 4:4:4, in this case in either *YUV* or *RGB* form. By using simple fixed ratios between the sampling rates, conversions between them can be achieved with relatively simple fixed-value interpolators. Apart from the obvious loss of bandwidth which must occur when initially changing to a lower sampling frequency, other conversion impairments, although tending to be cumulative, are minor. The hierarchy therefore allows systems to be developed for new applications in which 4:2:2 signals would be unsuitable, but still ensures that compatibility and straightforward signal conversions are retained.

Originally there was considerable interest in lower standards such as 2:1:1 or 3:1:0 (with the 1:0 notation indicating transmission of colour difference signals on alternate lines), primarily for application in low bit-rate transmission systems. Recently, however, low bit-rate systems interfacing directly with 4:2:2 signals have been developed and interest in lower members of the coding family has receded.

Over the same period, interest has grown in higher members of the family, with a number of proposals being made for high definition television standards. Clearly, it would be advantageous to make these compatible with the 4:2:2 standard in order to simplify conversions between the systems. However, because a high definition system may have more lines, a higher field rate and a wider aspect ratio, a system with simply related sampling frequencies, for example 16:8:8, will not necessarily imply a simple relationship of sample positions. In conversions involving higher scanning standards a simple relationship between the positions in the picture of sample points for the two systems is at least as important as a simple relationship between clock frequencies.
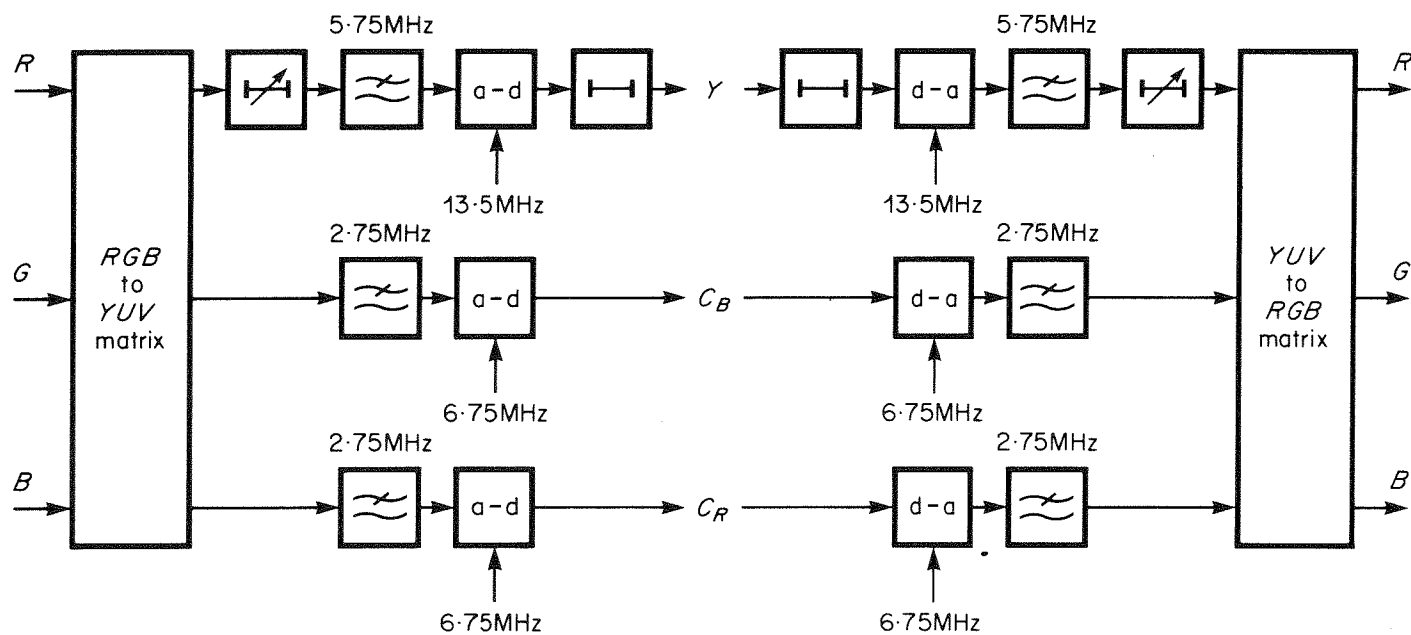
Two methods of converting between analogue *RGB* and 4:2:2 digital component signals have been used. The simpler method, shown in Fig. 3(a), uses analogue matrixing and filtering combined with the digital conversion of the component signals at 4:2:2 sample rates. The alternative method, shown in

Fig. 3(b), uses digital conversion of *RGB* signals, producing 4:4:4 *RGB* as an intermediate stage. As a result, the matrixing and some of the filtering is digital, giving more stable and better defined performance characteristics. The down-sampling process in Fig. 3(b) consists of low-pass filtering the 13.5 MHz colour-difference samples and then omitting alternate samples. For up-conversion, the 6.75 MHz samples are used directly, but interleaved with new alternate samples produced by interpolation.

3. MULTIPLE CONVERSIONS

Many parameters of the digital component standard have a direct influence on the conversion process from analogue *RGB*. These have been chosen

(a)
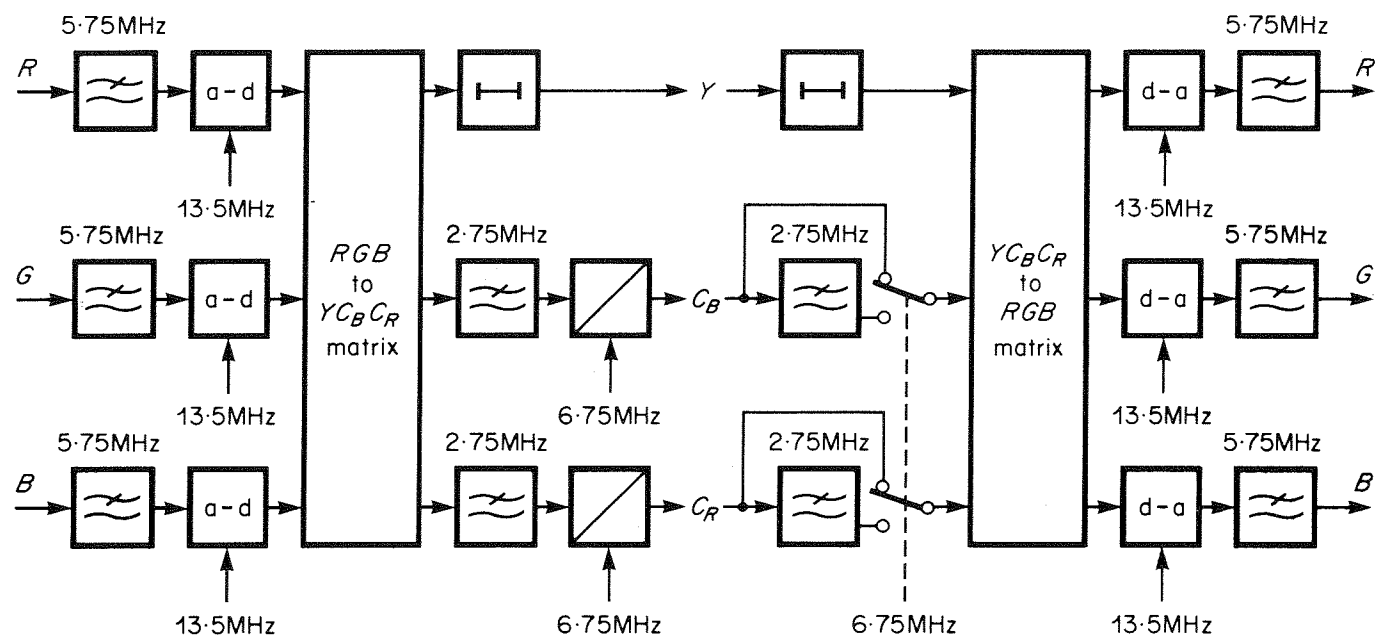
(b)

Fig. 3 - Methods of converting between analogue RGB and 4:2:2 digital components:
(a) using conversion at 4:2:2 sample rates and (b) using an intermediate 4:4:4 RGB stage.

to ensure that quality is maintained, even through multiple conversions.

3.1 Sampling structures

The digital coding standards of Recommendation 601 are based on line-locked sampling[7]. This produces a static orthogonal sampling grid in which samples on the current line fall directly beneath those on previous lines and fields, and exactly overlay samples on the previous picture. A pattern of orthogonal sampling sites is shown in Fig. 4(a).

Orthogonal sampling has many advantages for signal processing, such as the simplification of repetitive control waveforms, and, in terms of two-dimensional picture frequencies[12], supports a rectangu-

lar spectrum, as shown in Fig. 4(b). Compared with the quincunx[13] or line-offset structure, shown in Fig. 5(a), the rectangular spectrum retains the diagonal picture frequencies which are, in general, relatively unimportant. These can be sacrificed with quincunx sampling to obtain more horizontal resolution, as shown in Fig. 5(b). However, whereas a sharp-cut rectangular filter for orthogonal sampling is relatively easy to produce, a sharp response in two dimensions is much harder to achieve, requiring many lines of delay. Because of this, multiple conversions between analogue *RGB* and digital components or between different hierarchy levels would not have been practicable with a diagonally filtered quincunx system.

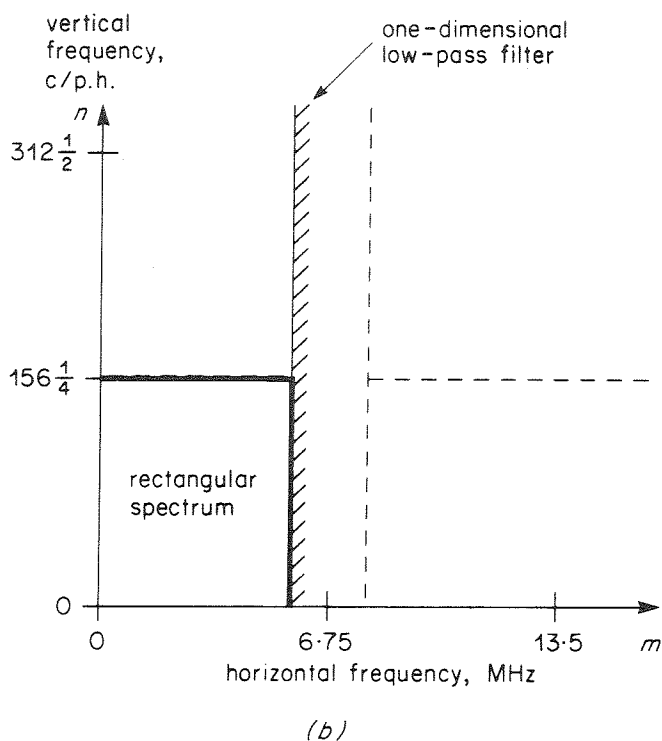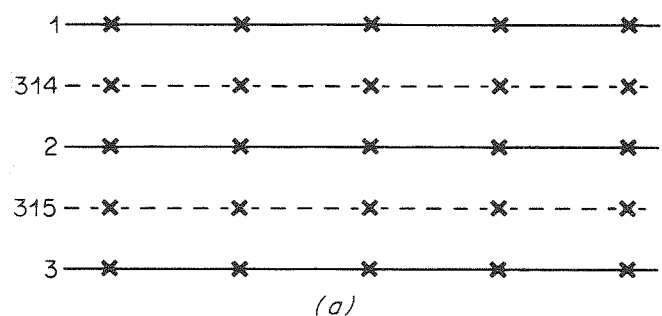Another important feature of Recommendation 601 is that, in addition to being locked in frequency,

Fig. 4 - Orthogonal sampling: (a) the position of sampling sites on the lines of an interlaced scanning raster (interlaced lines are shown dashed) and (b) the rectangular baseband region of the two-dimensional picture signal spectrum supported by the samples in (a). The one-dimensional filter cut-off returns the samples to continuous form.
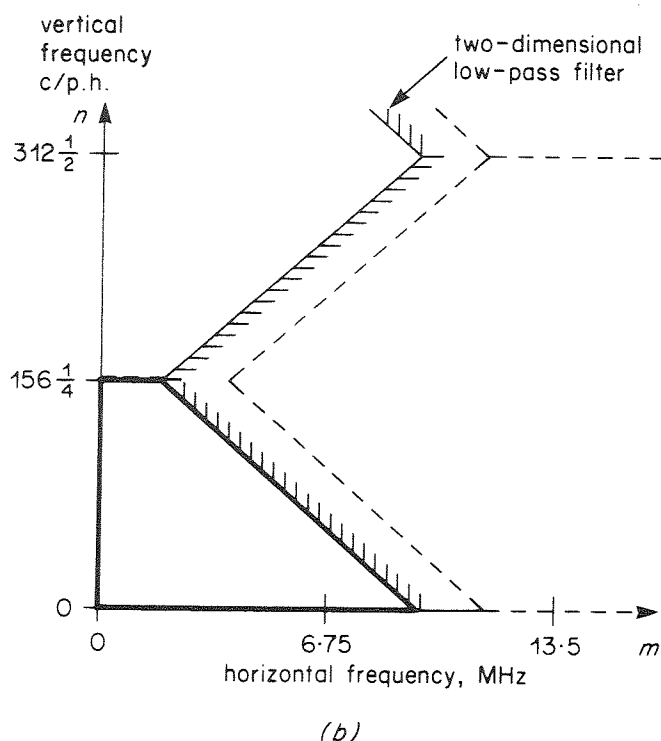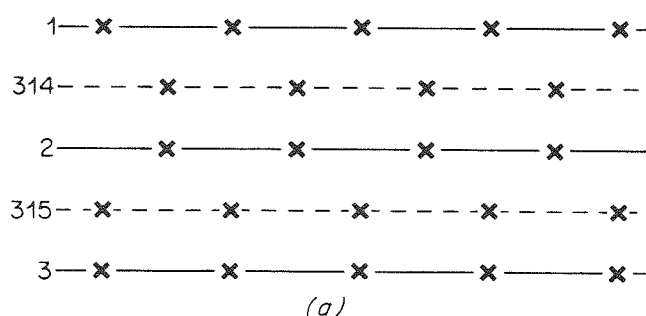

Fig. 5 - Quincunx sampling: (a) the position of sampling sites on the lines of an interlaced scanning raster (interlaced lines are shown dashed) and (b) the use of a two-dimensional filter allows more horizontal resolution to be retained, compared with Fig. 4.

the sampling is locked in phase so that one sample is coincident with the line timing reference point (the half-amplitude point of the falling edge of the line synchronising pulse) as shown in Fig. 6. This ensures that different sources produce samples at nominally the same positions in the picture. By making this feature common to all members of the sampling hierarchy, many of the samples from different levels of the hierarchy become co-sited, simplifying the conversion between different levels. This is shown in Fig. 6 for the lower rate colour-difference samples, known as C_B and C_R.

Setting the sampling frequency at 13.5 MHz provides a measure of commonality between 625/50 and 525/60 systems because this frequency has the property of being an exact harmonic of the line rate on both scanning standards. When the 13.5 MHz sampling frequency was first chosen, the potential problem of the 9th and 18th harmonics radiating to cause interference to the international distress frequencies[14] of 121.5 MHz and 243 MHz was not considered. It is believed, however, that interference can be avoided by careful equipment design[15].

3.2 Line formats

Further commonality between the 625/50 and 525/60 standards has been obtained by choosing similar digital line formats for the two standards, as shown in Fig. 7. With 13.5 MHz sampling, each digital line contains 864 or 858 samples, respectively, and consists of a blanking period followed by an active-line period. In each case, the analogue line timing reference falls in the blanking period, a few samples after the beginning of the digital line. Although the length of the full line is different for the two scanning standards, the same active-line period of 720 samples is used for both, in order to simplify dual-standard equipment and standards conversion. Samples at the beginning and end of the blanking period are set aside for timing reference codes, details of which are given in Section 5.2. With a sample rate of 6.75 MHz, as used for the C_B and C_R components of a 4:2:2 signal, each period shown in Fig. 7 contains half the number of samples. So, for example, there are 360 active-line samples and the full line contains 432 or 429 samples, depending on the scanning standard.

With analogue systems, difficulties sometimes occur with the repeated application of blanking causing an extension of the blanking periods and softening of blanking edges. With 720 samples, the digital active-line period is sufficiently long to accommodate the full range of analogue blanking tolerances on the two standards. There is, therefore, no
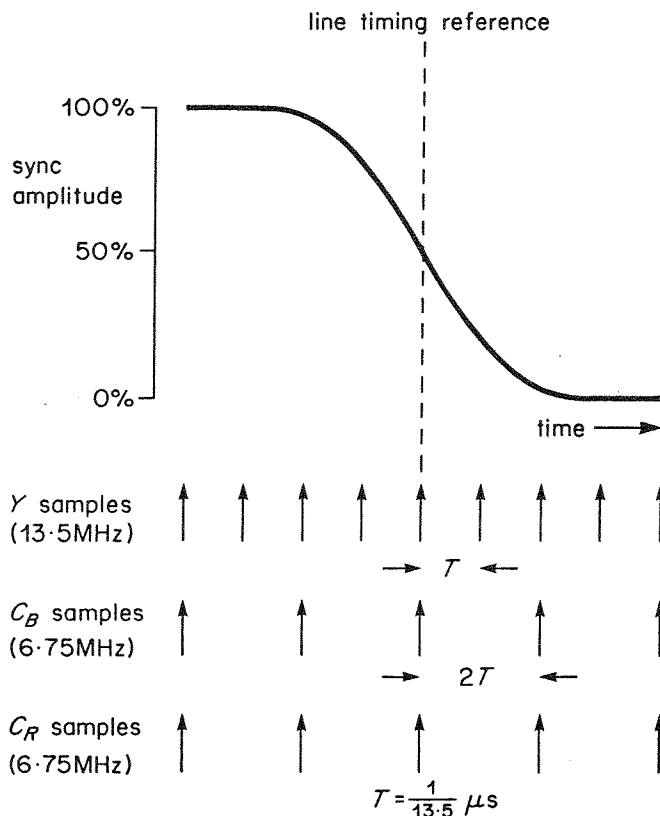
Fig. 6 - Positions of Y, C_B and C_R samples relative to the analogue line timing reference point.

difficulty with the repeated application of digital blanking. As a result, blanking to define the analogue picture width need only be applied once, preferably in picture monitors or at the conversion to composite signals for broadcast transmission. At this stage, blanking timings for the 625- or 525-line standards can be selected, as appropriate.

3.3 Quantisation

The 4:2:2 digital signals use eight bits per sample to convey each of the Y, C_B and C_R components. The eight-bit coding introduces a small amount of quantising distortion, which is signal dependent. In most circumstances, signals from television cameras and other picture sources contain sufficient noise to prevent the signal-dependent nature of the quantising distortion from becoming noticeable. Then the distortion appears noise-like and merely adds to the general noise level of the signal. However, if the signal is virtually noise-free, perhaps as a result of being generated electronically, then the quantising distortion can become noticeable as 'contouring' in areas where the signal brightness is changing gradually. This effect can be reduced significantly by the addition of half-sampling frequency dither at the analogue-to-digital (a-d) converter[1].

Fig. 7 - Digital line formats showing the number of 13.5 MHz samples in each section of the line:
(a) for 625/50 and (b) for 525/60 scanning.

An investigation carried out on multiple digital coding and decoding of PAL signals[16] has produced results which can be extrapolated to the lumin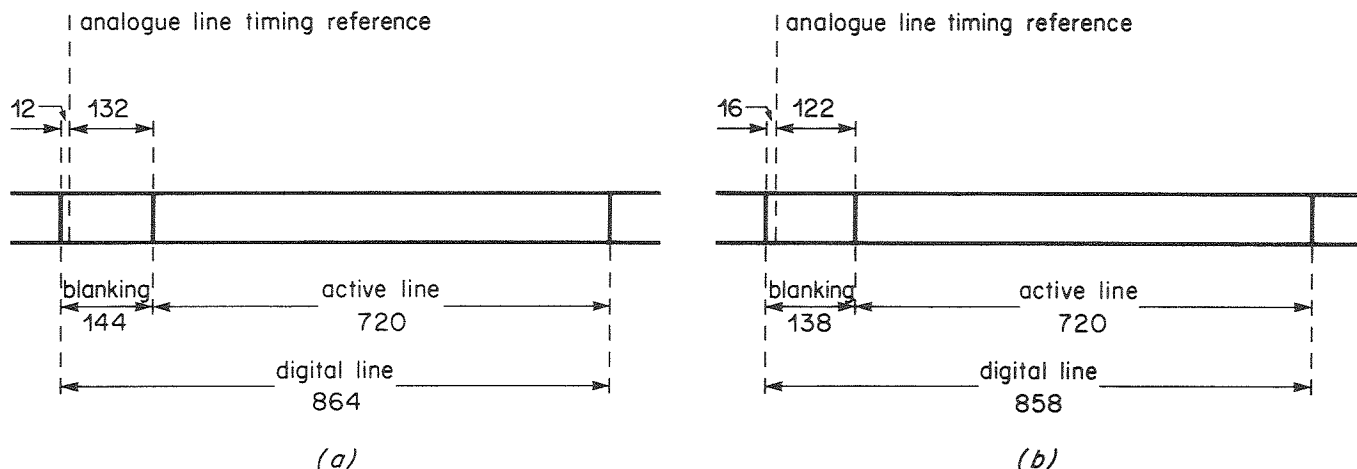ance component of a 4:2:2 signal. This showed that the increasing amount of noise caused by multiple quantisations was less significant than the effects of contouring and clamp streaking. It was concluded that, for up to eight codecs in tandem, impairments due to quantising luminance with eight bits per sample are very slight even on critical signals, provided that half-sampling frequency dither is used and attention is paid to clamp circuitry. Eight bits per sample is thought to be entirely adequate for the colour-difference signals, in which the impairments generally tend to be less noticeable than with luminance.

3.4 Luminance filtering

The use of digital sampling requires the bandwidth of the signal to be accurately defined by pre- and post-sampling lowpass filters, as shown in Fig. 8. As there may be many conversion processes in cascade, the filters used have to adhere to very tight tolerances to avoid a build-up of impairments[16]. The pre-sampling filter prevents aliasing, generally from low-level components, particularly noise, above the nominal video band. The post-sampling filter removes the high-level repeated spectra produced by the

sampling process. The sample-and-hold action of the digital-to-analogue (d-a) converter introduces a slow roll-off sin x/x characteristic which helps to suppress the repeated spectra, but also introduces a loss at higher in-band frequencies. This can be corrected conveniently by including a compensating lift in the nominally flat post-sampling filter response.

The CCIR template chosen for the characteristics of both the pre- and post-sampling luminance filters is shown in Fig. 9 (ignoring the sin x/x correction of the post-sampling filter). The amplitude characteristic is essentially flat to 5.75 MHz, with attenuations of at least 12 dB at 6.75 MHz and 40 dB for frequencies of 8 MHz and above. Details of the passband ripple and group delay specifications are summarised in Table 1. Both design limits and practical limits are specified in the table in recognition of the difficulty of accurately measuring such small deviations from ideal performance. The same template applies for RGB filters used in the conversion arrangement of Fig. 3(b).

During the development of the template, particular emphasis was placed on three of its features: the suppression of aliasing, the width of the passband and the flatness of the passband. Although aliasing is not a particularly disturbing impairment in high-frequency
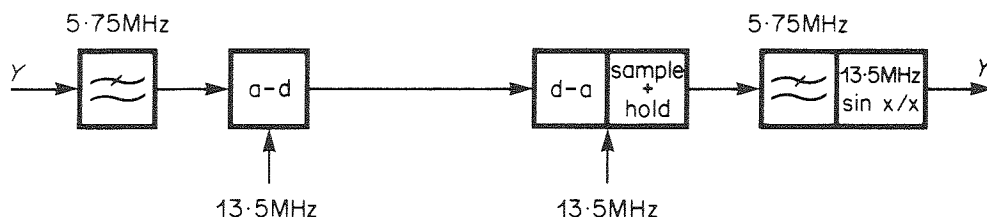

Fig. 8 - Filtering and sampling operations applied to the luminance signals in digital component
coding and decoding.

luminance, 12 dB attenuation at the half-sampling frequency ensures that it is kept at a very low level, even for out-of-band components generated within a digital process. Aliasing becomes even less of a problem when the signals encounter more than one codec.

The move to extend the bandwidth from 5.5 MHz, which resulted in a compromise figure of 5.75 MHz, made the transition band sharper still. Although the sharp-cut characteristic introduces an appreciable amount of ringing on luminance signal edges, the effect on picture quality is minimal, provided that group-delay performance is adequate. In the compromise, no corresponding change was made to retain the symmetry of the passband and stopband
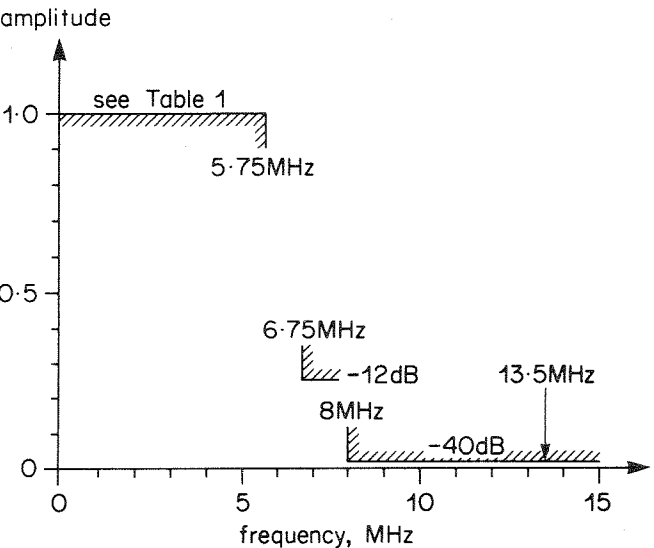

Fig. 9 - Filter amplitude characteristics template for the luminance component of a 4:2:2 digital signal. When used for analogue post-sampling conversions, it is assumed that sin x/x correction is included to give, overall, the passband response shown.

edges on either side of the half-sampling frequency. Because of this, the 5.5 to 5.75 MHz region may contain alias components which are not suppressed by the full stopband attenuation of 40 dB. In practice, however, the effect is unlikely to be significant.

When multiple codecs are considered, passband flatness and group-delay characteristics become increasingly important, particularly because with a single design of filter, the peaks of the passband ripples will be coincident. Unfortunately, the tight passband tolerances, coupled with the sharper transition band, make the template quite difficult to match. Because of this, there might be a temptation to relax the passband accuracy, which in any case is difficult to measure. However, the considerations above suggest that, if any relaxation is to be made, this would best be achieved by reducing the rate of cut, which would make a flat passband easier to realise.

3.5 Colour-difference filtering

While the considerations for filtering the colour-difference signals are, in most respects, broadly similar to those for luminance, differences have arisen. These have resulted partly from the lower colour-difference signal sampling frequency and partly from the possibility of using the two conversion methods shown in Fig. 3.

As with luminance, a flat passband with a sharp cut-off is required to provide a full range of frequencies for signal processing and to prevent repeated codings from producing a cumulative resolution loss. These features are particularly important for the colour-difference signals, to allow for downstream chroma-key[17]. However, the sharp-cut filters produce ringing at abrupt transitions which, because of the

Table 1
Luminance filter ripple and group delay tolerances.

Ripple:

Frequency range	Design limits	Practical limits
1 kHz to 1 MHz	±0.005 dB	increasing from ±0.01 to ±0.025 dB
1 MHz to 5.5 MHz	±0.005 dB	±0.025 dB
5.5 MHz to 5.75 MHz	±0.005 dB	±0.05 dB

Group delay:

Frequency range	Design limits	Practical limits
1 kHz to 5.75 MHz	0 increasing to ±2 ns	±1 ns increasing to ±3 ns

lower cut-off frequency, is much more noticeable than for luminance.

Ringing can be avoided by using filters with a slow roll-off (e.g. Gaussian) characteristic in the conversion from 4:2:2 signals. Therefore, while the process of conversion to 4:2:2 signals always uses sharp-cut filters, the conversion from 4:2:2 requires the option of using two different filter types. If subsequent processing is envisaged, such as when signals are returned to analogue *RGB* for interconnection to other digital component equipment, sharp-cut filters are used, whereas for the final conversion from 4:2:2, the slow roll-off response is also needed. With outputs to composite areas, the normal lowpass filters of the PAL encoder fulfil the slow roll-off function. By including similar filters at other appropriate outputs, such as for picture monitoring, ringing can be avoided while maintaining the full colour-difference bandwidth throughout the 4:2:2 signal area.

The series of filtering and sampling processes applied to the colour-difference components, corresponding to the two conversion approaches of Fig. 3, are shown in Fig. 10. In the direct 4:2:2 approach, shown in Fig. 10(a), the pre-sampling lowpass filter defines the colour-difference signal bandwidth. As with luminance, the characteristic of the post-sampling filter can conveniently include the sin x/x correction required to compensate for the sample-and-hold action of the d-a converter. The Gaussian filter response, where needed, could also be incorporated into a single characteristic. Because the lowpass characteristics are approximately a 2:1 scaled version of those for luminance, the analogue delays introduced are approximately double those in the luminance path. The luminance signal therefore requires compensating delays.

In the digital approach, Fig. 10(b), the signals undergo the 13.5 MHz sampling process as *RGB* signals. After matrixing, the colour-difference signals are lowpass filtered and reduced to a rate of 6.75 MHz by omitting alternate samples. Using a transversal filter has the advantage of providing a perfect linear phase characteristic, eliminating the need for group-delay correction. Interpolation for the up-conversion process to 13.5 MHz introduces no sample-and-hold loss, so sin x/x correction is not required in this case. The Gaussian characteristic, where needed, could be incorporated into the interpolator characteristic or provided separately in the 13.5 MHz signal after interpolation.

The template for colour-difference filters is shown in Fig. 11 with the response essentially flat to 2.75 MHz. However, as aliasing is less noticeable in the colour-difference signals, the attenuation at the half
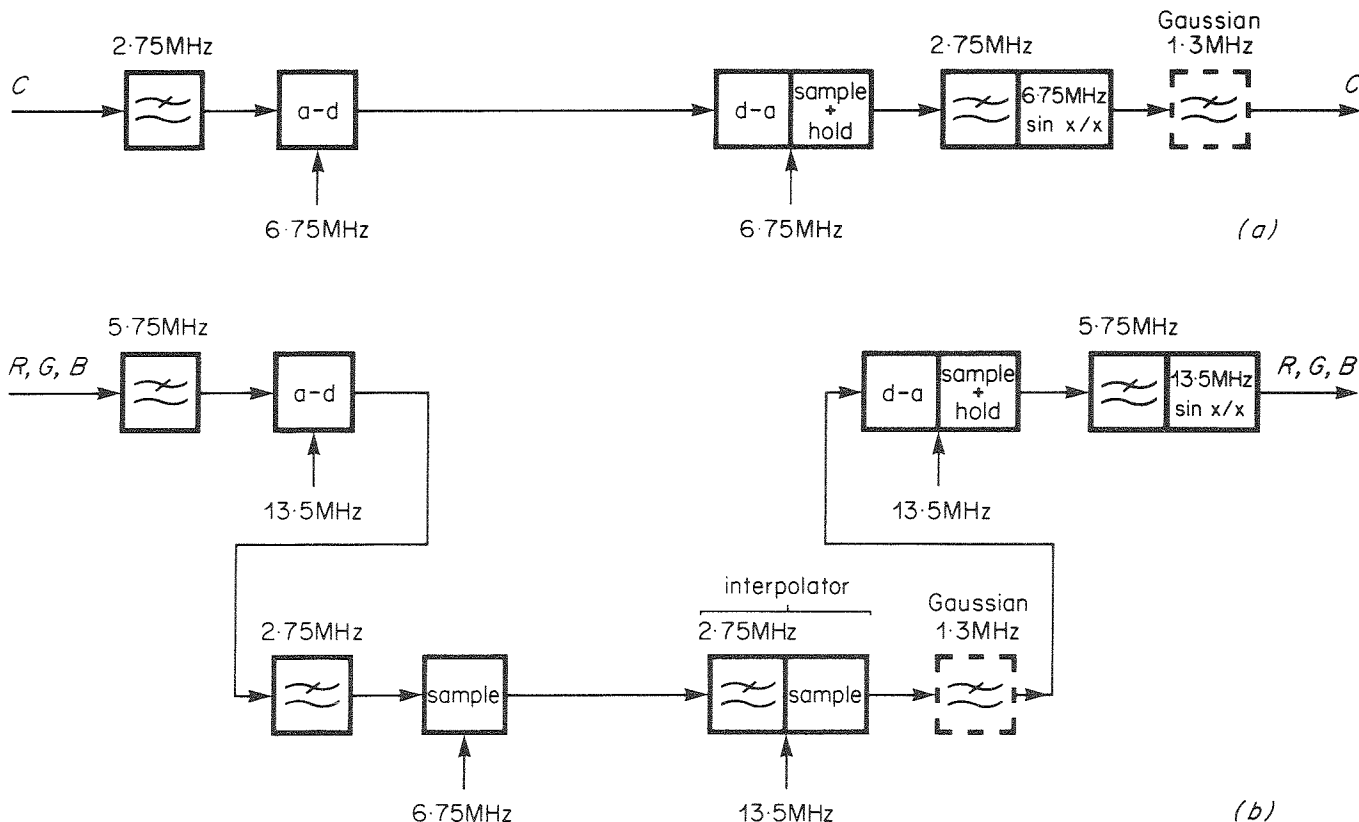

Fig. 10 - Filtering and sampling operations applied to the colour-difference signals in digital component coding and decoding: (a) with direct conversion to 4:2:2 sample rates, and (b) with an intermediate conversion to 4:4:4 RGB.
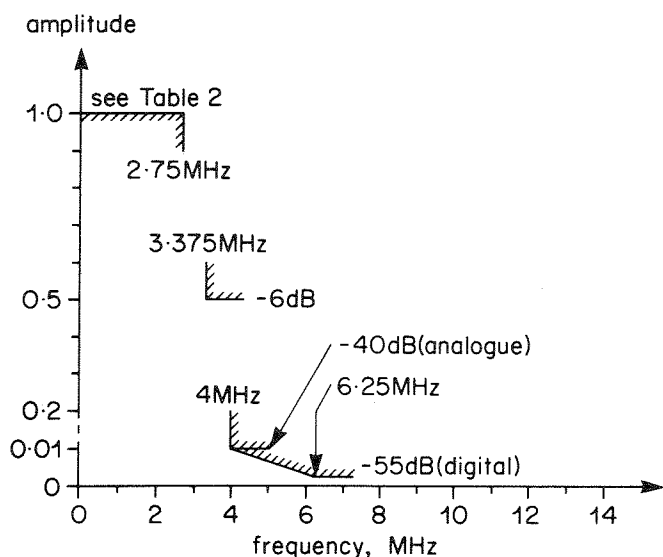
amplitude

Fig. 11 - Filter amplitude characteristic templates for the colour-difference components of a 4:2:2 digital signal. When used for analogue post-sampling conversions, it is assumed that sin x/x correction is included to give, overall, the passband response shown. A greater stopband attenuation is specified for digital post-sampling filters because no sin x/x attenuation occurs.

sampling frequency is relaxed to 6 dB. For digital filters there is a particular advantage in using a skew-symmetric response passing through the −6 dB point at the half sampling frequency as this makes alternate coefficients in the filter zero, thus almost halving the number of taps. This allows the use of an efficient technique[18,19], in which a single filter is used for both the C_B and C_R signals. Also for digital filters, the

attenuation requirement in the stopband is increased to 55 dB because there is no attenuation from sin x/x sample-and-hold loss. When using a skew-symmetric filter, this constraint is met automatically because of the tight passband tolerance. Passband ripple and group delay tolerances are shown in Table 2.

The Gaussian characteristic shown in Fig. 12 is similar to those specified for colour-difference filters in composite colour encoders, having an attenuation of approximately 6 dB at 2 MHz. This is sufficient to prevent ringing altogether.

amplitude
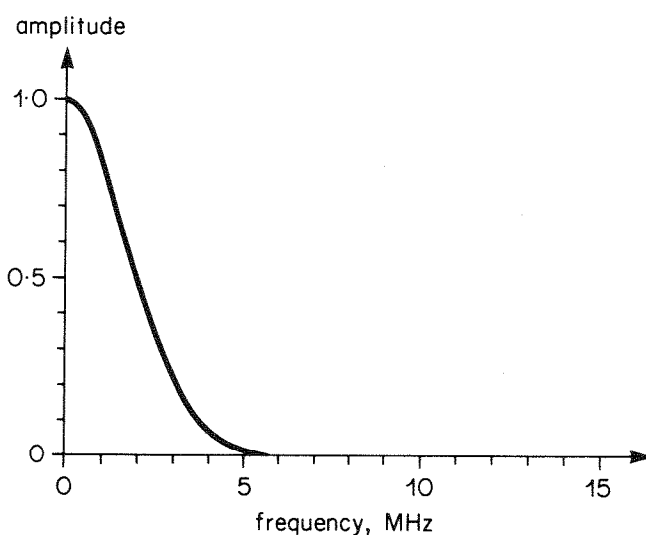
Fig. 12 - A Gaussian slow roll-off colour-difference filter characteristic suitable for preventing ringing at the final output of a 4:2:2 digital system.

Table 2
Colour-difference filter ripple and group delay tolerances.

Ripple:

Frequency range	Design limits	Practical limits
1 kHz to 1 MHz	±0.01 dB	increasing from ±0.01 to ±0.05 dB
1 MHz to 2.75 MHz	±0.01 dB	±0.05 dB

Group delay:

Frequency range	Design limits	Practical limits
1 kHz to 2.75 MHz	increasing from 0 to ±4 ns	increasing from ±2 to ±6 ns
2.75 MHz to f_{-3dB}	—	±12 ns

Note: The use of transversal digital filters ensures zero group delay distortion.

4. SIGNAL PROCESSING

4.1 Coding ranges

The choice of coding ranges for digital component signals balances the requirements of providing adequate capacity for signals beyond the normal range and minimising quantising distortion. Although black level is reasonably well defined, white level can be subject to some variation in camera pictures. Even with a standard level signal, the addition of noise, or transients produced by sharp-cut filtering can both take the signal outside the nominal range. Signal processing can sometimes produce signal components extending well beyond the normal range, but allowing sufficient headroom for such signals would lead to a noticeably higher level of quantising distortion. However, a small margin beyond the nominal black and white levels can provide for overshoots and minor gain variations, without significantly affecting the signal-to-quantising noise ratio.

The eight-bit coding of Recommendation 601 corresponds to a range of 256 levels, referred to as 0 to 255 in decimal form or 00 to FF as hexadecimal codes. Levels 0 and 255 are reserved for timing reference information (detailed in Section 5.2), leaving levels 1 to 254 for signal values. While the initial proposals suggested using equal symmetrical coding ranges for all three components, after discussion the standard incorporated a greater margin for overloads at the white end of the luminance range. This reflected the observation that impairments due to limiting at white level were slightly more visible than at black. Thus, a standard level luminance signal extends from black at 16 to white at 235, while the colour-difference signals occupy a symmetrical range about a zero chrominance level of 128, extending down to level 16 and up to level 240. The coding levels for the Y, C_B and C_R signals corresponding to a 100% colour bars waveform[20] are shown in Fig. 13. The coding range used for 4:4:4 digital RGB signals is the same as that for luminance, as this simplifies the matrixing process in digital conversions between RGB and YC_BC_R signals.

Additional gain factors have to be used to normalise the ranges of the B-Y and R-Y signals. With a luminance range of 0 to 1, the ranges of B-Y and R-Y are -0.886 to 0.886 and -0.701 to 0.701, respectively. The C'_B and C'_R signals are made to occupy an equal range to luminance (although in this case -0.5 to 0.5, rather than 0 to 1) by multiplying the B-Y and R-Y signals by conversion factors, as follows:

$$C'_B = \frac{0.5}{0.886} (B\text{-}Y)$$

$$= 0.564 \ (B\text{-}Y)$$

and

$$C'_R = \frac{0.5}{0.701} (R\text{-}Y)$$

$$= 0.713 \ (R\text{-}Y)$$

In digital conversions of the form shown in Fig. 3(b), it is also necessary to take account of the factor 224/219 to allow for the difference in the coding ranges of luminance and colour-difference signals.

4.2 Number systems

The binary coding ranges allocated to the Y, C_B and C_R signals in Fig. 13 provide an efficient
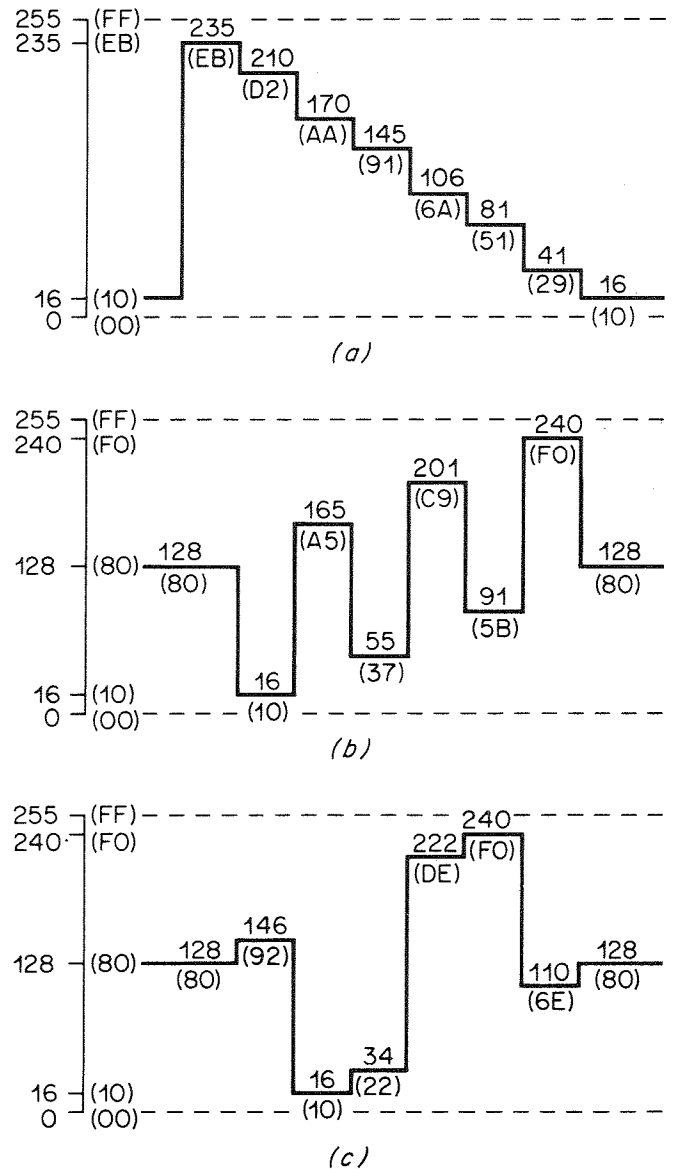

Fig. 13 - Coding levels in a 4:2:2 digital component 100% colour bars signal:
(a) Y, (b) C_B and (c) C_R components. Numbers in parentheses show the codes in hexadecimal form. Because of the limited resolution of an eight-bit digital signal, the coding levels shown only give an approximation to the ideal waveforms.

method of signal representation for applications such as distribution and storage, but can present some problems for signal processing. Signal processing often involves binary arithmetic, such as addition, subtraction, multiplication or, less frequently, division. This may introduce negative numbers, such as for negative coefficients in a digital filter or to represent the results of a subtraction. Also, the signals themselves can take positive and negative values. Luminance is essentially a positive quantity, but, although negative brightness has no physical meaning, small negative signal values can occur as transients or as the result of noise. In contrast, the colour-difference signals naturally produce a wide range of positive and negative values.

Using binary numbers to represent negative quantities as shown in Fig. 13 is known as offset binary coding. Difficulties arise because the zero of the signal is offset from the zero of the binary coding range as shown in Table 3. For example, if the offset binary values are multiplied by a factor, the offset is also multiplied, which causes an unwanted shift of the signal zero position. Offset binary represention therefore requires a suitable correction to be added to each answer to return the offset to its original value. However, general purpose integrated circuits for digital arithmetic, such as arithmetic logic units and multipliers, tend to use a different method for representing negative numbers, known as the two's complement code. As shown in Table 3, this uses the normal binary representation for positive values and zero signal is represented directly by the zero code. Negative numbers are represented by the two's complement (the bit-by-bit inversion of the magnitude plus one least significant bit). In practice, for signals

with a centre zero such as the colour-difference signals, the two's complement codes are formed by merely inverting the most significant bit of each binary number in the coding range.

With luminance, the signal zero (black level) is at level 16 rather than in the middle of the coding range. Setting binary zero at black level would therefore require nine bits with a two's complement code because of the larger range of positive excursions. However, a more efficient alternative is to treat luminance as a signal with roughly equal positive and negative excursions from a zero value near mid-grey. This makes the codes for luminance signals similar to those for the colour-difference signals, increasing the amount of circuitry that can be made common to all three signals. Even a process such as fading to black, which depends on the position of the zero, can be accomplished conveniently by implementing a cross-fade, with black level set as the final value.

The sign and magnitude representation, also listed in Table 3, also has special properties which are sometimes useful for signal processing. In this case, only the sign bit (the most significant bit) alters between the positive and negative values of a number. In addition to the obvious advantages in applications where symmetrical positive and negative signals are involved, sign and magnitude coding gives an increased tolerance to bit errors, particularly for the colour-difference signals. However, conversion from the signal coding ranges of Fig. 13 is relatively complicated.

The alteration to the luminance coding range to provide increased overload capacity for signals beyond white has resulted in a coding range with an even number of levels between black and white. As a consequence, there is no level corresponding exactly to a mid-range signal, such as may be required in the digital synthesis of test waveforms. An approximation to the correct level can be provided by alternating between the adjacent levels, 125 and 126, although this is not aesthetically very satisfactory. This and other drawbacks outweigh the marginal benefit from the increased overload range at the white end of the luminance signal.

4.3 Range extension

While the coding ranges shown in Fig. 13 provide a small margin for overloads, much larger signals can be produced by digital signal processing. This can occur, for example, when two signals are combined together or if a filter has greater than unity gain at any frequency. The generation of numbers outside the coding range results in number overflow; this produces errors equal to the full coding range,

Table 3
Representation of negative numbers: correspondence between signal levels and code values for the offset binary, two's complement and sign and magnitude codes.

signal value	offset binary	two's complement	sign and magnitude
+127	1111 1111	0111 1111	0111 1111
.	. .	. .	. .
.	. .	. .	. .
.	. .	. .	. .
+1	1000 0001	0000 0001	0000 0001
0	1000 0000	0000 0000	0000 0000
−1	0111 1111	1111 1111	1000 0001
.	. .	. .	. .
.	. .	. .	. .
−127	0000 0001	1000 0001	1111 1111
−128	0000 0000	1000 0000	— —

causing very obvious picture impairments. This is normally avoided' by adding more significant bits to extend the coding range so that the largest signals envisaged can still be accommodated.

Simply adding a more significant bit to the binary coding range only increases the range in the positive direction, as shown in Fig. 14(a). It is
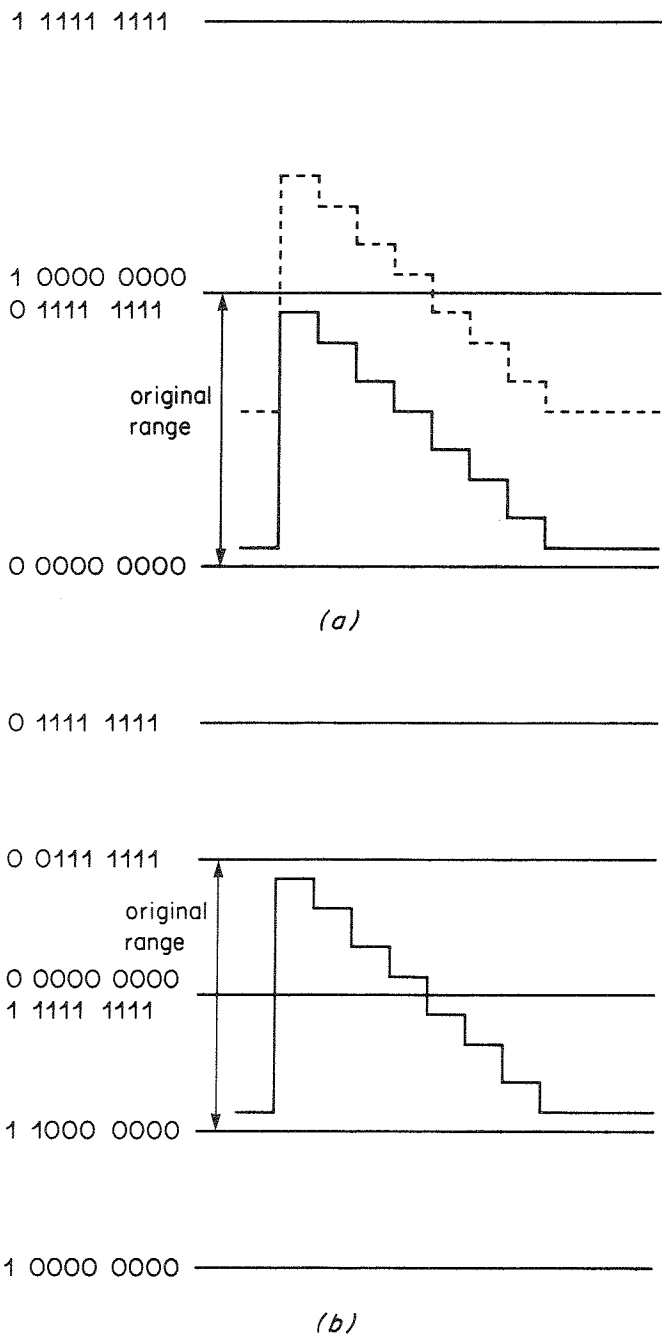
therefore necessary to shift the original signal into the middle of this new range by adding 128. However, with the two's complement code, adding an extra bit increases the range symmetrically, without the need to modify the existing bit values. This is shown in Fig. 14(b). The two's complement coding range can be extended as necessary by repeating the most significant bit value of existing sample words into higher bit positions.

Some processes can generate less significant bits which provide increased coding precision. Discarding bits below the original eight at each stage would lead to a build-up of rounding errors, causing an increase in quantising noise. Because of this, less significant bits are frequently retained within a process, then reduced to eight bits only when necessary, such as at the output of a unit.

4.4 Rounding

After signal processing it is usually necessary to return the signal words to eight-bit form. Although this is bound to result in a less accurate representation of the signal, the errors can be minimised by rounding and limiting the values before the bits are discarded.

The effect of truncating the word values for a linear ramp signal is shown in Fig. 15(a). Each signal word takes the next eight-bit value below the actual value, so that the truncation introduces a bias towards the negative end of the coding range. Although the effect is small in eight-bit signals, this can accumulate over several truncations to produce a noticeable shift. In luminance, this is a shift towards black and so produces a slightly darker picture, but in the colour-difference signals the shift is towards negative U and V, so introducing a slight green cast to the picture.

The errors of truncation can be reduced by rounding, that is, before truncation, adding 1 at the level one bit below the lowest bit to be retained (equivalent to half the least significant bit). Each signal word then takes the nearest eight-bit value to the actual value, as shown in Fig. 15(b). For unquantised signals, this eliminates the bias, resulting in an equal distribution of positive and negative errors. However, signal processing may produce only a small number of additional bits. For example, with two extra bits there would be additional levels at the ¼, ½ and ¾ positions. While those at the ¼ level would be rounded down, the ½ and ¾ positions would both be rounded up, thus introducing a slight positive bias. In the colour-difference signals, this can be avoided by rounding depending on the sign of the signal. Accordingly, the ½ level would be rounded up for positive values and rounded down for negative values. This technique preserves the symmetry of the colour-

Fig. 14 - The effect of adding more significant bits to the coding range of a digital signal:
(a) when using offset binary coding and (b) when using two's complement coding. With the offset binary code, it is necessary to shift signal values back to the middle of the coding range in order to preserve the zero position.
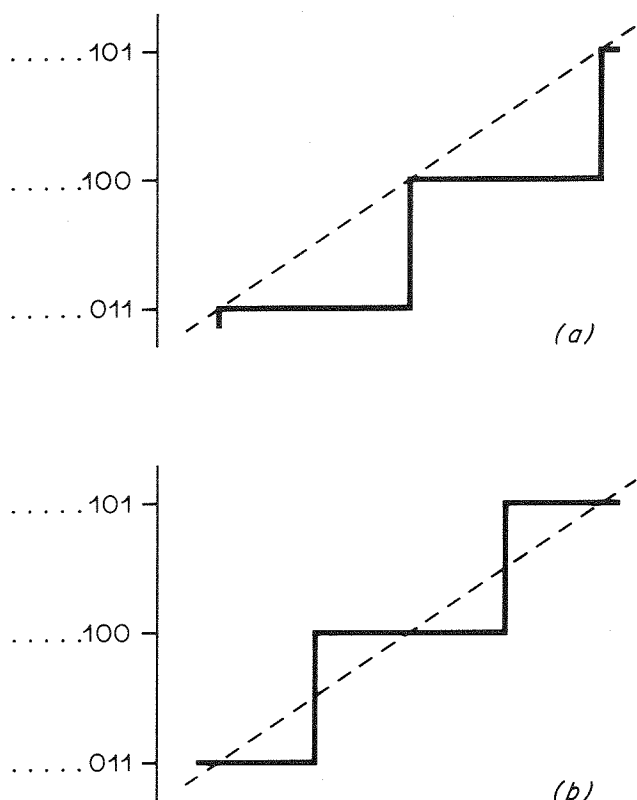
Fig. 15 - The effects of removing bits of lower significance than the original eight bits:
(a) by truncation and (b) by rounding. The true signal is shown dashed and the quantised signal is the solid line in each case.

difference signals and prevents the introduction of a colour cast.

Although rounding minimises the errors due to truncation, a further reduction in error visibility can be obtained using the error feedback technique[21]. Instead of discarding the residual bits, these are accumulated from one sample to the next and so influence the truncation of subsequent samples. This process breaks up the highly visible coherent low frequency quantisation distortion and replaces it by much less visible high frequency components. Nevertheless, it is still advantageous to keep to a minimum the number of occasions on which truncation to eight bits occurs.

4.5 Limiting

Before discarding bits with higher significance than the original eight, it is necessary to detect any values outside the normal 1 to 254 range and to substitute a value within the range. Substituting the value 254 for positive overloads and 1 for negative overloads minimises the occurrence of limiting, although using values further within the overload range would avoid a concentration of sample values adjacent to the timing reference code values, 0 and 255.

The process of limiting produces harmonics of the fundamental signal frequencies. In analogue signals this is not particularly serious because many of the harmonics are beyond the normal signal bandwidth and tend to be removed by subsequent filtering operations. However, the sampled nature of digital signals causes the harmonics to be aliased to occupy much more noticeable low frequencies. With transients, limiting reduces the overshoots and consequently the resulting impairments. In this case, the alias components are not noticeable as a separate feature from the transient. In contrast, applying limiting to frequency gratings can produce noticeable aliasing, even with relatively small overloads. In this case, the most visible patterning is produced by frequencies near to one quarter of the sampling frequency. Therefore, it is preferable to avoid limiting in the sampled domain, for example by ensuring that signals conform reasonably well to the standard level before digital encoding.

The headroom at each end of the coding range, although small, is sufficient to avoid the need to limit directly at the black and white levels. This is advantageous because the limiting of noise near black can cause a black shift due to rectification, whilst at white this appears as 'patchy' noise as the result of aliasing.

When signals are generated or processed in the form of digital *YUV*, there is the possibility that signals could be produced for which the corresponding *RGB* values extend far beyond their normal ranges. Such signals can be obtained merely as a result of band-limiting the colour-difference signals for 4:2:2 sampling. The presence of invalid *YUV* combinations is not immediately apparent because of the inter-dependence of their magnitude limits. For example, when the luminance signal is at black or white, any non-zero values of the colour-difference components will result in invalid values of *R*, *G* or *B*. Because of this, a technique has been developed[22] for limiting *YUV* signals to ensure that the corresponding *RGB* limits are observed. To minimise the impairment, the limiting process maintains the luminance and hue of the original signals, altering only the saturation to bring the signals within their prescribed limits.

5. INTERCONNECTIONS

An important step in the introduction of digital component signals occurs when the 'islands' of digital processing are joined by digital interconnections. It is at this point that several major problems have to be faced. In addition to conveying the digital signals between units or areas, there are the necessities of selecting sources with digital switching matrices and synchronising the timing of different sources.

5.1 Multiplexing

For convenience of distribution, the Y, C_B and C_R samples of the 4:2:2 signal are formed into a single 27 MHz multiplex in the order C_B, Y, C_R, Y, as shown in Fig. 16. The alternative of grouping all the Y samples, then all the C_B samples and finally all the C_R samples together, although more convenient for monitoring, was dismissed on economic grounds. The stream of data words corresponding to each line always begins with a C_B sample. In the multiplexed sequence, the co-sited samples (those that correspond to the same point on the picture) are grouped as C_B, Y, C_R. Unused samples in the digital blanking intervals are set to blanking level, that is, 16 (10 hex) for Y, and 128 (80 hex) for C_B and C_R. Codes marking the end of active video (EAV) and the start of active video (SAV) occupy, respectively, the first four and last four sample positions of the digital blanking interval.

Interconnection methods for 4:4:4 digital signals are much less developed than for 4:2:2. However, proposals are being discussed for conveying a 4:4:4 *RGB* signal in two 27 MHz multiplexes, so that routeing equipment already developed for 4:2:2 can be used. The resulting 4:4:4:4 arrangement would include capacity for a 13.5 MHz linear key signal for use in chroma-key applications. The proposed format of the key signal would be similar to that of luminance with levels 235 and 16 representing the wanted and unwanted portions, respectively, of the picture.

5.2 Timing reference codes

The four-word SAV and EAV codes, shown in Fig. 16, allow the start and end positions of the active-line samples to be identified easily. The first three words use the reserved codes 00 and FF (hexadecimal) to form a preamble sequence FF, 00, 00. This marks the position of the hexadecimal line label word known as XY. In binary form, X has values 1FVH, with F giving odd/even field information, V denoting vertical blanking and H differentiating between EAV and SAV codes. The values of F, V and H on different lines are listed in Table 4. The Y portion of the XY word provides a four-bit parity check $P_3P_2P_1P_0$ to protect the X data, as shown in Table 5. The parity bits P_3,P_2,P_1,P_0 are defined by the following relationships in terms of F,V and H:

$$P_3 = V \oplus H$$

$$P_2 = F \oplus H$$

$$P_1 = F \oplus V$$

$$P_0 = F \oplus V \oplus H$$

where $\oplus$ represents modulo-2 addition (the exclusive-OR function).

The F, V and H bit values produce odd/even field, vertical blanking and horizontal blanking waveforms directly. The generation of other waveforms requires the use of counters reset by line and picture pulses. The detection of a line pulse is reasonably easy and unlikely to be affected by errors, with missed codes compensated by the flywheel action of the counter. However, obtaining a pulse to reset the picture counter is less secure, as this has to be generated by decoding the change of state from 1 to 0 in the F value. Thus, if the code containing the first 0
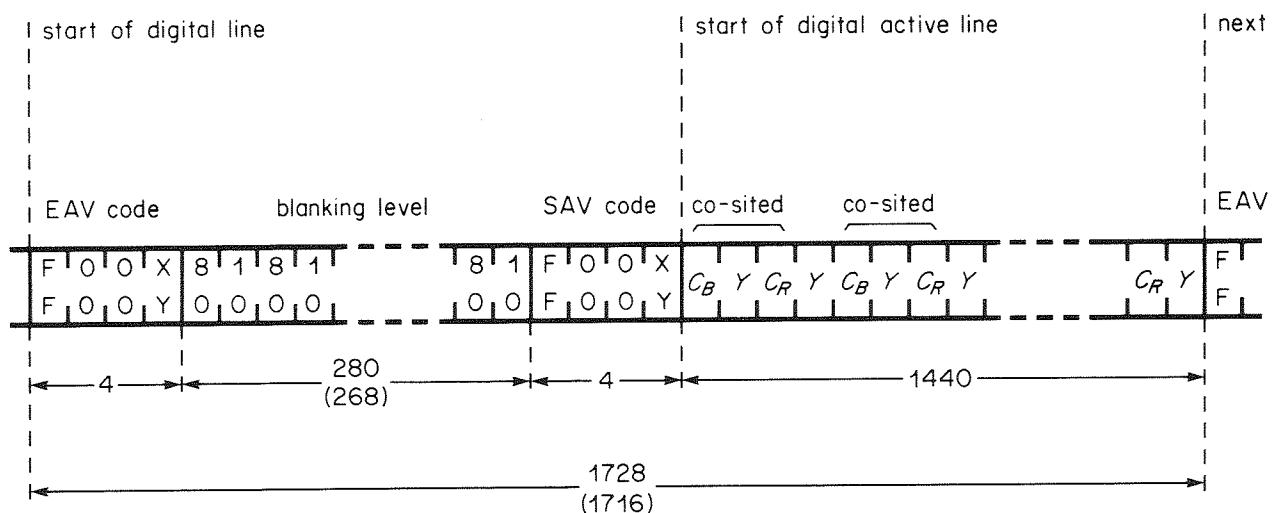

Fig. 16 - Data format for multiplexed 4:2:2 video signals for the 625/50 scanning standard. Sample values are shown as hexadecimal numbers, while the numbers of samples are shown in decimal form. Where different, the values for 525/60 scanning are shown in parentheses.

Table 4
Timing reference code values.

625/50 line numbers	F	V	H(EAV)	H(SAV)
1-22	0	1	1	0
23-310	0	0	1	0
311-312	0	1	1	0
313-335	1	1	1	0
336-623	1	0	1	0
624-625	1	1	1	0

525/60* line numbers	F	V	H(EAV)	H(SAV)
1-3	1	1	1	0
4-9	0	1	1	0
10-263	0	0	1	0
264-265	0	1	1	0
266-272	1	1	1	0
273-525	1	0	1	0

* Line numbers are in accordance with new engineering practice for 525/60 in which numbering starts from the first line of equalising pulses, instead of the first line of broad pulses as had previously been used.

Table 5
Parity values for timing reference codes.

	BINARY						HEXADECIMAL
	X			Y			XY
1	F	V	H	P_3	P_2	P_1	P_0	

1	F	V	H	P_3	P_2	P_1	P_0	XY
1	0	0	0	0	0	0	0	80
1	0	0	1	1	1	0	1	9D
1	0	1	0	1	0	1	1	AB
1	0	1	1	0	1	1	0	B6
1	1	0	0	0	1	1	1	C7
1	1	0	1	1	0	1	0	DA
1	1	1	0	1	1	0	0	EC
1	1	1	1	0	0	0	1	F1

value of F is missed, the change in F is delayed and the counter will be reset to the wrong position. An alternative is to sense changes in the value of V and to gate these with the F waveform, but this technique is no more secure. Fortunately, in a studio environment, the protection provided by the four-bit parity check should be more than adequate. However, if the FVH codes are to be used in more extreme environments, a more secure method of conveying picture synchronising information would be advantageous. The number of samples between the EAV and/or SAV codes can be counted to distinguish between 525- and 625-line signals.

The beginning of the EAV code marks the beginning of the digital line period, which starts slightly before the corresponding analogue line. This avoids having two periods of blanking associated with each digital line. The start of the digital field occurs at the start of the digital line which includes the start of the corresponding analogue field. Each digital field, therefore, contains a whole number of digital lines. The half lines of active picture in the analogue signal are accommodated as full lines in the digital signal.

When demultiplexed into 4:2:2 Y, C_B and C_R signals, the line code words are distributed automatically to give, in hexadecimal form: 00, XY in the Y signal, FF in the C_B signal and 00 in the C_R signal. This proves convenient because the luminance signal carries the main synchronising information in the XY word, with the 00 word acting as a preceding marker. In the colour-difference signals, the FF and 00 codes provide timing markers for synchronisation and act as labels to distinguish between the C_B and C_R signals. When converted into the form of RGB 4:4:4 signals, the 00, XY codes from luminance carry through into all three signals, provided that the contributions from the colour-difference signals are suppressed during matrixing.

5.3 Parallel interface

In the parallel interface format[8], the multiplexed 8-bit samples and a 27 MHz clock signal are conveyed down nine pairs in a multi-way cable with 25-way D-type pin connectors (male) at each end. The individual bits are labelled Data 0-7, with Data 7 being the most significant bit. The pin allocations for the individual signals are listed in Table 6. The two spare pairs of contacts are reserved for bits of lower significance than Data 0. Equipment inputs and outputs both use sockets (female), so that interconnection cables can be used in either direction.

Signals on the parallel interface use logic levels compatible with the balanced drivers and receivers of Emitter-Coupled Logic. With the NRZ data format, this allows transmission over distances of 50 m unequalised or 200 m with equalisation. The rising edge of the essentially square, 27 MHz clock waveform is the active edge and, at the sending end, is timed to occur in the middle of the bit-cell of the data waveforms. As the higher frequency clock signal tends to suffer a greater delay than the data, coincidence of

Table 6
Pin connections in the 25-way parallel interface.

1	Clock A	14	Clock B
2	System ground	15	System ground
3	Data 7A	16	Data 7B
4	Data 6A	17	Data 6B
5	Data 5A	18	Data 5B
6	Data 4A	19	Data 4B
7	Data 3A	20	Data 3B
8	Data 2A	21	Data 2B
9	Data 1A	22	Data 1B
10	Data 0A	23	Data 0B
11	Spare A-A	24	Spare A-B
12	Spare B-A	25	Spare B-B
13	Cable shield		

The notation A and B is used to denote the two terminals of a balanced pair. For a logic 0, A is negative with respect to B, and, for a logic 1, A is positive with respect to B.

the clock and data transitions is usually the factor that limits the distance for reliable transmission.

5.4 Serial interface

Conversion to the serial interface format allows the signals of the 27 MHz multiplex to be conveyed down a single bearer, either coaxial cable or optical fibre. This is achieved by converting the eight-bit samples to a nine-bit format according to an adaptive mapped encoding procedure. The nine-bit values are transmitted as NRZ data, least significant bit first, to produce a 243 Mbit/s data stream. The eight to nine-bit conversion process includes extra coding to allow reliable clock recovery and word synchronisation[8].

For coaxial connections, 75 Ω cables with BNC connectors are used. A specification for optical fibre interfaces has yet to be agreed, although this approach appears to hold the best prospects for long-distance transmission.

A serial approach is also favoured for routeing and switching digital signals, primarily because of the reduced number of bearers compared with the parallel interface. A particularly attractive approach, currently under intensive development, is to use a single optical fibre carrying many tens or even hundreds of digital signals[23]. The technique used to carry the signals, known as WTDM, is a combination of optical wavelength-division multiplexing (WDM) and electrical time-division multiplexing (TDM). Because all the signals are distributed on a single fibre, the system combines the routeing and switching functions to achieve a very flexible, high performance system.

6. STUDIO SYNCHRONISATION

6.1 Handling of timing reference codes

The presence of the timing reference codes described in Section 5.2, although forming part of the multiplexed digital component signal, often results in the need for additional processing. This can affect both the code sequence and the signal sample words from the point at which the signal is first encoded, through any signal processing operations, until the signal is returned to analogue form. These additional operations are shown in general terms in Fig. 17.

At the source encoder, Fig. 17(a), the appropriate FVH information has to be generated from the incoming mixed sync waveform and inserted at the correct positions in the digital data stream. In addition, any occurrences of the reserved codes (00 and FF) in the signal data have to be erased, being replaced by the adjacent codes (01 and FE). Although it is unlikely that an exact imitation of a full FF, 00, 00, XY sequence will occur, there is no guarantee that subsequent circuitry will check all aspects of the code.

Whenever a signal with timing reference codes is processed, the operations shown in Fig. 17(b) may be required. Before the process, it is necessary to separate the codes into a by-pass path. This prevents the process corrupting the codes. Also, the codes should be blanked before processing to prevent operations such as filtering from spreading the codes into adjacent active-line sample positions. At the output, any reserved codes in the processed signal must be erased before reinserting the codes from the by-pass path, appropriately timed to take account of the delay introduced by the process.

Returning the signals to analogue form, such as to display the picture on a monitor, involves the timing reference code operations shown in Fig. 17(c). The FVH information must be extracted and used to regenerate the analogue synchronising and blanking waveforms. If for any reason analogue blanking is not applied, the codes must be blanked from the signal path to prevent interference with the clamp operation of the monitor.

6.2 Central distribution

While the operations described above are appropriate for individual items of digital component equipment connected in tandem, when more complicated arrangements are used, other factors have to be taken into account. In Fig. 17, the 27 MHz clock associated with the multiplexed signals is handed on from one piece of equipment to the next. Similarly, the synchronising information is taken from the input

and any processing delays are accommodated by delaying the timing reference codes to match. This approach becomes less satisfactory as more digital equipment is connected together.

One problem is that the stability of the clocks tends to be degraded by passing through inter-connecting cables and through equipment. At the extreme, this could become sufficiently poor that reliable clocking could not be maintained. Even before this, the jitter could be sufficient to affect the quality of a subsequent composite encoding process[24]. Further problems arise when it is necessary to combine two or more digital signals together, for example, at a mixer. Then, at some point, all the signals have to use a common clock and the sync timing of the individual signals has to be aligned.

Under these circumstances, it would be convenient to have some form of common reference signal, carrying clock and synchronising information. The timing of all the signals could then be aligned approximately by distributing this signal in parallel to each piece of equipment. The minor differences in timing that would result from differing cable delays could be accommodated by short-range synchronisers at the combining point. For remote sources, synchronisers with the capacity to store a picture period of the incoming signal would allow perfect alignment. The principle of this arrangement is shown in Fig. 18.

6.3 Synchronising signals

Several proposals have been made on the form that the synchronising signal should take, ranging from
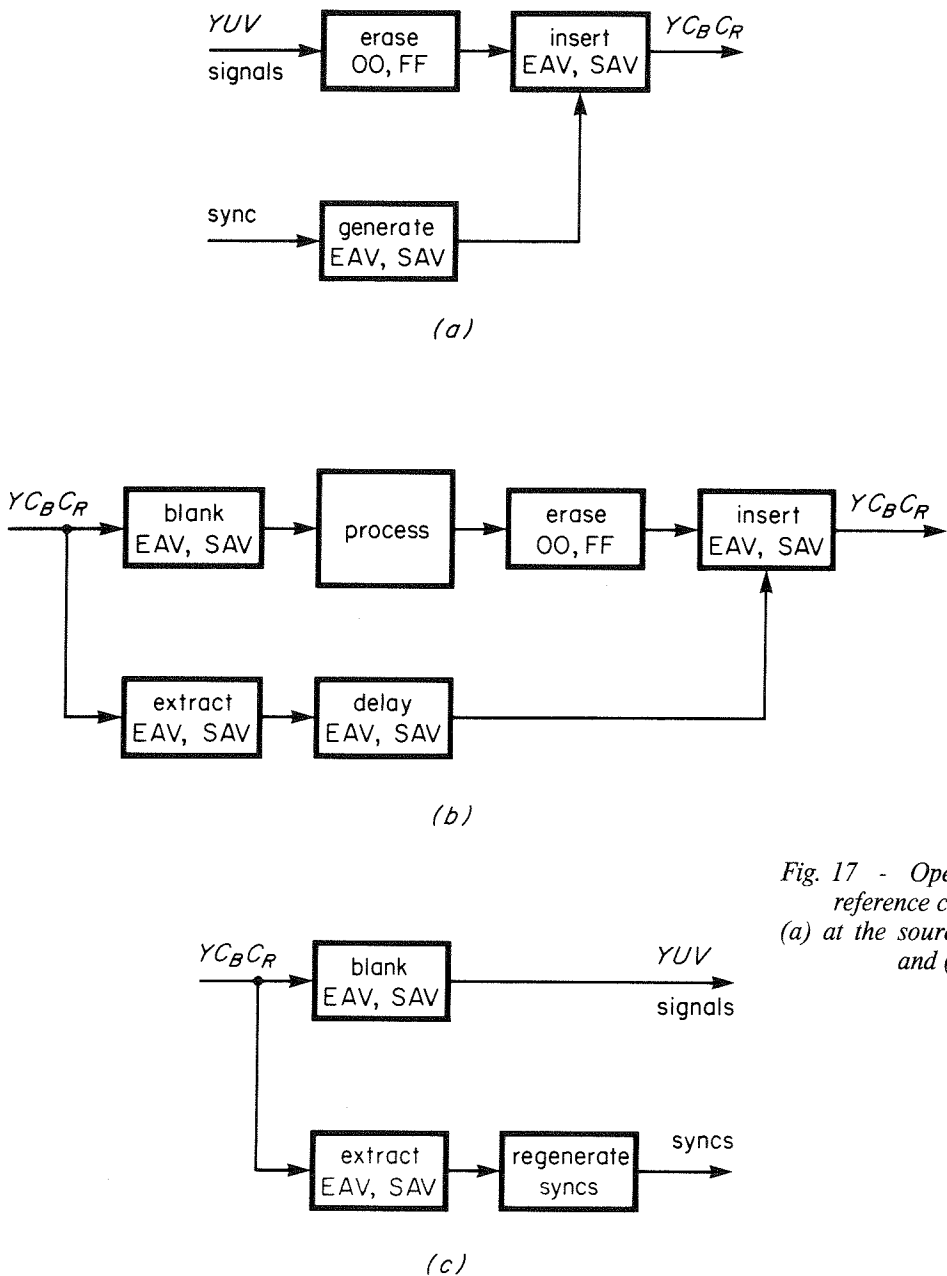

Fig. 17 - Operations concerned with the timing reference codes in a digital component signal: (a) at the source coder, (b) during signal processing and (c) at the display conversion.

A working reference is then required, against which the action of the equipment under test can be compared. Also, simple signal path errors, such as bits stuck at 1 or 0, or shorted together, will be revealed anyway, simply by checking the timing reference codes. The scope for applying specialised digital test sequences is therefore rather limited.

7.2 Analogue waveforms

The strength of analogue signal waveforms, albeit synthesised directly as digital samples, is that they produce an easily recognisable output when converted to analogue form. For example, the 100% colour bar waveforms, shown in Fig. 13, are frequently used for analogue testing. In the context of digital component signals, the waveforms exercise the full range of values of all three signals together, both in terms of 4:2:2 YC_BC_R and 4:4:4 RGB signals. Also the extremes of the standard level coding range can be identified easily for checking signal levels. However, the highly structured form of the colour bar signal results in a very limited number of word combinations, which may fail to reveal the presence of errors.

In this respect, sawtooth or ramp waveforms are an improvement, because virtually all the levels are used. For luminance, the dual ramp waveform,
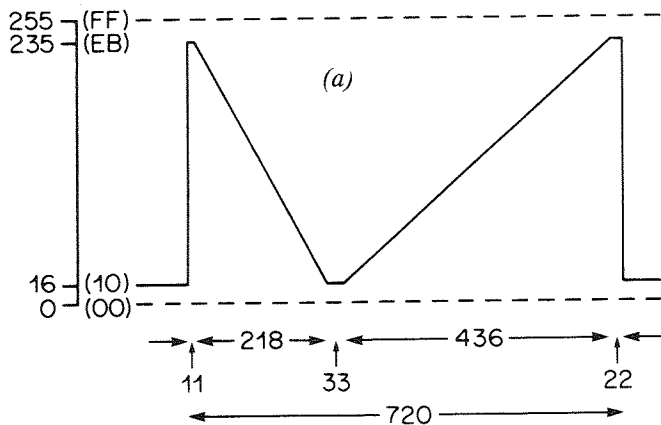
shown in Fig. 19(a), has some advantages. First, the slope of each ramp has been chosen to produce a constant number of samples per step. This causes both the waveforms of the individual bits and the quantising distortion in the resulting analogue signal to assume regular patterns. Irregularities introduced by subsequent circuitry are therefore more noticeable. In addition, using the reversed ramp at the beginning of the line ensures a large signal at each end of the digital line, so that the effect of any blanking processes can be assessed. Finally, the flat portions of the waveform identify the black and white positions in a standard level signal. The waveform produced on an oscilloscope after d-a conversion is shown in Fig. 19(b).

The luminance waveform of Fig. 19(a) can also be used for R, G and B signals. When originated in RGB form, the combined signal can be used to test luminance circuits. Alternatively, by suppressing some of the RGB signals, non-zero colour-difference signals can be produced to test the C_B and C_R channels. A shortcoming of this method is that the colour-difference signals so produced do not extend over the full coding range. In this respect, then, the colour-difference waveform shown in Fig. 19(c) is preferable and provides similar advantages to the waveform of Fig. 19(a). However, in legitimate signals, the maximum values of C_B and C_R can only occur when
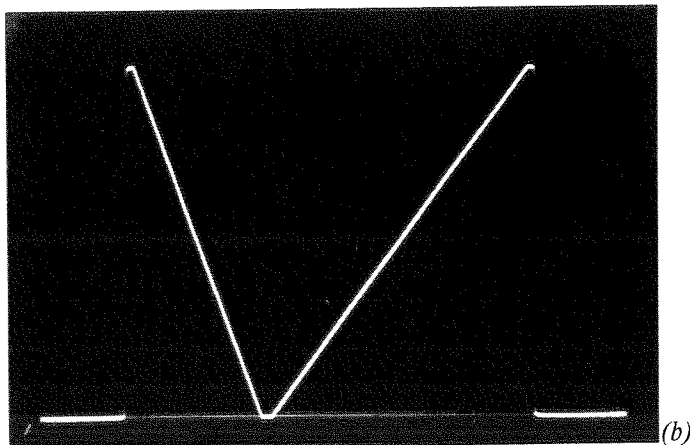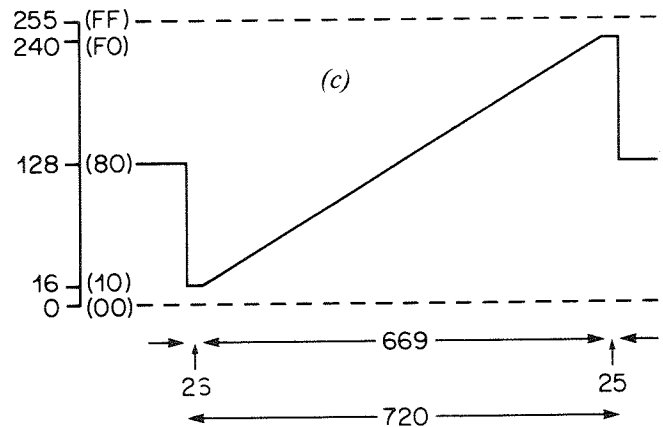

(a)

(c)

(b)

Fig. 19 - Ramp test signals for use in 4:2:2 digital component systems:
(a) for luminance, (b) the oscilloscope waveform produced by the signal in (a), and (c) for the colour-difference channels.

a standard analogue mixed sync waveform to a digital 27 MHz multiplexed signal with the active-line video at blanking level. Between these two, the variations include a signal comprising a basic clock waveform, either at 27 MHz or a sub-multiple such as 4.5 MHz, modulated with FVH data by binary phase-shift keying.

Each of these proposals represents a different set of compromises. Mixed syncs is simple to distribute and is universally available in analogue studios. However, the generation of jitter-free 27 MHz clocks and the derivation of FVH information from mixed syncs are in neither case trivial tasks. Also the relevance of analogue mixed syncs in a digital studio environment is questionable. In a fully digital environment it would therefore be more convenient to use a 27 MHz multiplexed signal for the reference, as this provides the clock and timing reference codes directly in the required form. While this may be more difficult to distribute, at least initially, this does allow the possibility of locking equipment directly to a digital signal source as the reference, when circumstances require it.

A fully digital environment, however, may not be the norm for several years and until then there will be a mixture of analogue and digital equipment. In recognition of this, the EBU has recommended[25] that digital equipment should be able to accept a 300 mV mixed sync waveform, optionally with a colour burst, as a reference signal for synchronising information. Even so, this does not preclude the provision of a digital synchronising input as an alternative.

7. TEST SIGNALS

Proposals for test signals for digital component systems have been broadly of two types: digital codes and analogue signal waveforms.

7.1 Digital sequences

Digital codes, such as pseudo-random binary sequences, can be used as the basis for automatic test equipment to check for errors in paths that should be transparent, for example, digital stores or distribution links. Such equipment is particularly necessary when the errors only occur as the result of complex bit sequences. However, if, as a result of signal processing, the digital equipment makes any legitimate alteration to the bit stream, the technique is more difficult to use.
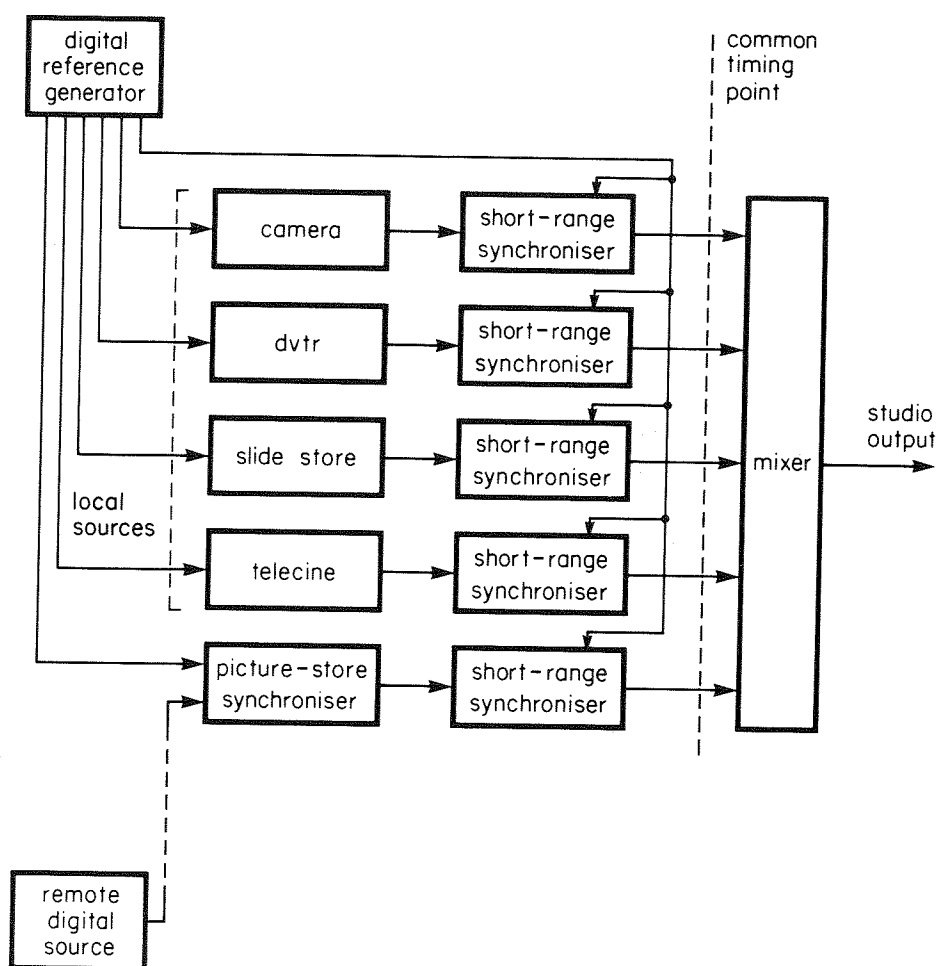

Fig. 18 - Synchronising arrangements in a digital component studio.

the corresponding luminance value is at mid-range, so the waveforms of Fig. 19(a) and (c) cannot be used together. Fig. 19(c) could be used legitimately by setting the luminance to a mid-range value, except that, as mentioned in Section 4.2, no mid-range luminance level exists. An unfortunate consequence, therefore, of the white level modification to increase the headroom for overloads is that the standard is unable to support a valid set of full-range C_B and C_R values.

The ramp waveforms, although exercising more levels than the colour bar signal, still represent a relatively limited set of word combinations, particularly as most of the signal content is low frequency. For some applications, for example filter testing, linear sweep waveforms, as shown in Fig. 20, are more suitable. By relating the rate of sweep to the line duration, the waveform can be calibrated in frequency with reasonable accuracy. Thus filter characteristics can be displayed directly and impairments such as aliasing revealed. For general applications, the more varied word sequences of the sweep waveforms are more likely to provoke errors than the more structured colour bar and ramp waveforms. As with the ramp waveforms, sweeps can be generated in RGB or YC_BC_R form, with the same coding range restrictions applying.

As the waveforms discussed above are all line repetitive, the most convenient method of generation is to read the values out of a read-only memory. Sample values for band-limited edges can be calculated by a technique described elsewhere[5]. When multi-dimensional signal processing is used, it is less feasible to store appropriate test waveforms and direct generation of the signals becomes more efficient.

Electronic zone plate test signals[26] prove useful in a particularly wide range of applications.

8. CONCLUSIONS

The digital component studio standards described in CCIR Recommendation 601 reflect a number of assumptions about the way in which digital signals would be used in studios. In this Report, several features of the digital standards have been described, with a view to explaining the assumptions made and the reasoning used to support their choice. An understanding of these factors is required if the full potential of the signals for picture quality and operational flexibility is to be maintained through multiple signal conversions and in complex signal processing operations.

Some features of the standard are still the subject of discussion and development. To add to these discussions, some of the factors involved in signal synchronisation in a digital studio are further explored. This appears to suggest that for a fully digital environment, the use of synchronising signals in the form of normal 27 MHz multiplexed digital signals would be advantageous. Some suggestions for the form of test signals to be used in digital equipment are also made.

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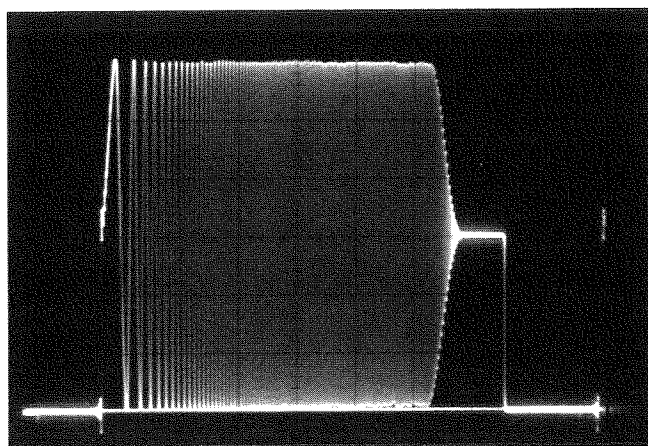
(a)
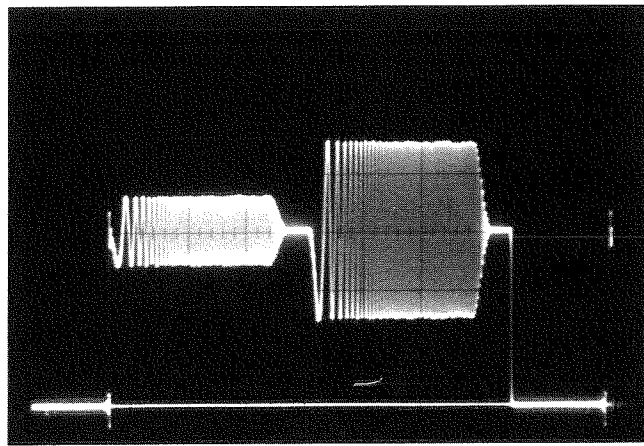
(b)

Fig. 20 - Oscilloscope waveforms of sweep test signals for use in 4:2:2 digital component systems:
(a) for luminance and (b) for the colour-difference channels. The waveform in (b) shows the green signal produced by successive full-amplitude C_B and C_R sweep signals extending to 5.5 MHz.

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